User Manual

S03 Series

IP Phone

Version 1.1

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Introduction

About This Manual

This Manual provides basic information on how to install and connect S03 IP Phone to the network. It also includes features and functions of S03 IP phone components, and how to use them correctly. We sincerely hope you could enjoy the convenience and capabilities brought forward by our products.

Before Getting Started

Before you can connect S03 to the network and use it, you must have a high-speed Internet connection installed. A high-speed connection includes environments such as XDSL, cable modem, and a leased line.

S03 IP Phone

S03 IP phone is a stand-alone device, which requires no PC to make Internet calls. S03 IP phone supports both data and voice thru IP network, and also provides all the features and functionalities of conventional phone and more.
Our IP phone guarantees clear and reliable voice quality on IP network, which is fully compatible with SIP industry standard and able to interoperate with many other SIP compliant devices and software on the market.

Notice

This publication describes the instruction for S03 series IP phone functions only. We reserve the rights to do any changes or make enhancements of this publication without further notice. The most updated electronic revision of user manual can be downloaded from WISH's website: www.wishproduct.com timely, thanks for your understanding and continuous support.

Installation

2.1 Appearance

S03 series IP phone are designed to look like the conventional telephones. The following photo illustrates the appearance of S03 IP phone.

2.2 Package list

1) One S03 IP phone (Main body + Handset + Cord)
2) One Straight Ethernet cable
3) One universal power adapter
4) One User Manual

Warning: Although the adapter of S03 series IP Phone is compliant with UL standard, please do not attempt to use other difference power adapter or cut off power supply during configuration or updating phone. Using other power adapter may damage S03 series phone and will void the manufacturer warranty.

Caution: All operations of our product must abide by the instructions provided by IP Phone manufacturer. Any Changes or modifications to this product without formal authorization by manufacturer, or operation of S03 series phone in any way other than the instructions stated on this user manual will void the manufacturer warranty.

2.3 Installation

1) Insert handset cord into the handset jack and left jack of IP Phone
2) Insert the power adapter's plug into the phone front Power jack (DC 5V) and the 2-prong plug end of which into grounded power outlet
3) Remove the LAN cable for Internet connection from your PC and connect it to WAN port of S03, then follow below installation checking way

2.4 Check Installation by LATTICE of LCD

S03 IP Phone has a 75mmx24mm LCD that can display three lines of below characters each. Here is the display when all segments illuminate:
The LCD is equipped with a backlight. When the phone is in the normal idle state, the backlight is off. Whenever an event occurs, the backlight turns on automatically and brings the user's attention. The definitions for each character displayed on LCD described as below table.

Product Overview

S03 IP Phone is a next generation IP network telephone based on industry open standard SIP (Session Initiation Protocol) and H.323.Built on innovative technology, S03 series IP Phone offer customer superb sound quality and rich functionalities at mass-affordable price.

3.1 Software Features

 Support two modes: Bridge and Router (NAT&NAPT)
 Network Protocols: TCP/UDP/IP, ICMP, HTTP, DHCP Client （ WAN Interface）, DHCP Server（LAN Interface）, DNS Client， DNS Relay， SNTP, FTP， TFTP
 VOIP Protocols: Support H323 (V4) & SIP (RFC3261, RFC3262, RFC3264, RFC3265) synchronously
 Voice Codecs: G.711（A-law/U-law）, G.723.1(High/low)， G.729
 NAT transversal: Support STUN client, AVS and Citron etc. Can modify SIP register port, HTTP server port， Telnet server port and RTP port
 Support two SIP server synchronously：Can register two different SIP Server, and can make a call by either proxy
 Support standard voice features such as numeric Caller ID Display, Call Waiting, Hold, ransfer, Do-Not-disturb, Forward, in-band and out-of-band DTMF, Hotline (off-hook autodial), auto answer, ban outgoing
 Full duplex hand free speakerphone, redial, call log, volume control, voice record with indicator
 Support standard encryption and authentication (DIGEST using MD5, MD5-sess)
 Support Silence Suppression, VAD (Voice Activity Detection), CNG (Comfort Noise eneration), Line Echo Cancellation (G.168), and AGC (Automatic Gain Control)
 Provide easy configuration thru manual operation (phone keypad， Web interface and Telenet) or automated centralized configuration file via TFTP or HTTP
 Support firmware upgrade via TFTP/FTP and HTTP
 Support syslog, can send event of phone to syslog server

3.2 Hardware Specifications

The below table describes the hardware specifications of S03 Phone

Item S03

Power Adapter Input: 100-240VAC 50~60Hz
Output: +5VDC, 500mA
CPU DSP621 PSB21553 150MHz
WAN 10/100Base T RJ-45 for LAN Port
Power Consumption Idle: 1.4W / Active: 1.8W
LCD size 65×24mm
Operating Temperature 0～40℃(32°～104℉)
Storage Temperature -10°～60℃(14°～140℉)
Relative Humidity 10～65% (Non-condensing)
Dimension (W×H×D) 20.5×22.5×7.5cm
Weight (packaging included) 0.80kg

Certification CE / FCC Part 15 Class B

BASIC OPERATIONS

4.1 Get Familiar with Keypad

S03 phone has a 25-button keypad. Definitions of each state as below

Key Button Mode Definitions

Key Button	Mode	Definitions
0 -9	In the dial-up mode	Decimal digit number 0-9, star and pound keys are usually used to make phone calls
	In the keypad configure mode	Rapid first button press display the digit number 0-9, rapid second button press display the English character or others
*	In the dial-up mode	As one part of phone number when call out
	In the call hold mode	Ready to call a third party’s number to make three-way (or conference) calling
	In the keypad configure mode	Equal to the dots notation when input IP address
#	In the dial-up mode	As one part of phone number if “#” is the first dialed number, otherwise as the ending symbol to end up dialing
DISPLAY	In the IDLE mode	Continuous thrice press display assigned IP address of WAN Port, gateway’s IP address and the phone number registered on public server
ENTER	In the keypad configure mode	Confirm configuration or enter submenu mode
HOLD	In the keypad configure ode	Temporarily hold the active call
REDIAL	In the dial-up mode	Redial the number last dialed, or force a call to go out immediately before timeout
H.F	In the IDLE mode	Enter hands-free mode
UP	In the keypad configure mode	Go back to previous menu item or increase handset/speakerphone volume
DOWN	In the keypad configure mode	Go down to lower menu items or reduce handset/speakerphone volume
XFER	In the keypad configure mode	In the keypad configure mode

4.2 Dialing or Making Calls

4.2.1 Dialing general PSTN Phone or extension.

There are three dialing modes:

1. Pick up handset or press SPEAKER button, and then enter the phone numbers, IP phone will

send out the numbers with the DTMF tones.
2. In the off hook mode or Hands-free mode, Press the REDIAL button directly to redial the number last called. Once pressed, the last dialed number will be displayed on the LCD with DTMF tones and an outgoing call is sent.
3. Make use of Speed dialing mode: enter PBOOK→SPEED DIAL→INPUT INDEX submenu to input the index which correspond to the phone number you want to dial, then the phone could automatically dial the number.

4.2.2 IP to IP calling

Making IP to IP calling is nearly same as dialing general PSTN Phone, there are three ways to set IP phone number and domain
1. "Peer to Peer" calling mode: direct make calls and no need to set phone number thru proxy server (user could refer to Dial peer setting on web configuration charter).The phone should be operated under following condition (satisfy one option)
 Both two phones are assigned the public IP address individually
 Both two phones using private IP address should be on the same LAN.
2. Dialing the IP number configured by public agency, both two phones should be already registered on the public server.
3. Dialing the IP number configured by private agency, both two phones should be already registered on the private server.

Examples:

To dial a number on the proxy, such as 1001, simply pick up handset or press speakerphone, dial 1001 and then press the REDIAL button.
To dial a PSTN number such as 83429400, you might need to enter in some prefix number followed by the phone number. Please check with your VoIP service provider to get the information. If your phone is assigned with a PSTN-like number such as 83429400, most likely you just follow the rule to dial 83429400 as if you were calling from a regular analog phone, followed by pressing the REDIAL button.

4.3 Answering Calls

There are three ways to receive incoming calls:
 Pick up handset to receive incoming calls
 Press the H.F button to receive incoming calls
 Start-up the voice message function (Press REC button) to record the incoming calls, afterwards listen to voice messages thru phone audio play
 Operations: Enter REC→RECEIVED→NEW→LIST NO submenu
Note: User can switch from a hands-free call to handset by picking up the handset. To switch from a handset call to a hands-free call, press hands-free button, and then hang up the handset.

4.4 Call hold

Pressing the HOLD button during current conversation enables you put an active calling on hold temporarily while a second call is answered or made, press this button again will go back to the previous call.

4.5 Call transfer

Press the XFER button enables users could transfer an incoming call to transfer the incoming call to the third party's number. When both sides are on conversation, users could inform the called person to announce the call will be transferred and then press the XFER button, dial another telephone number when you hear the dial tone, then the incoming call will be transferred to another line and users conclude the previous telephone conversation.

4.6 Three-Way Calling

S03 phone support three-way (or conference) Calling. That is users could talk to more than one person (up to two) at the same time.

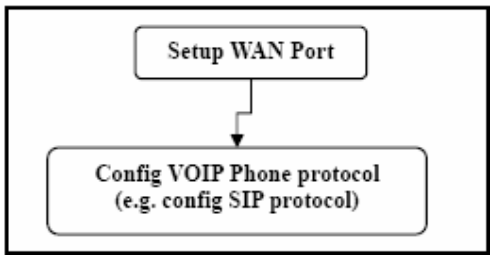
Process: press CONF or HOLD button →Dial third party's number->put through→ press HOLD button again

Once the three-way initiator concludes the three-way calling, the other two sides can not continue the conversation call and hand up automatically.

Configuration Guide

5.1 Config IP Phone through Keypad

Below is the configuration procedure for basic operations**:**



5.1.1 Minimum configuration

1. Network configuration:
    1. Host Configuration
        1. IP Configuration
            1. IP Address
        2. MAC Address
        3. Net Mask Configuration
            1. Net Mask Address
        4. Host Name
        5. Domain Name
            1. Domain Name
            2. Domain Name Se
        6. DHCP
            1. Enable
            2. Disable

2. Router Configuration
    1. Gateway Address
3. DNS Server
4. Call Manager

3. Password
    1. Enter Password

4. Reboot
    1. Reboot Now
    2. Reboot Later

Press the button select list item

Press"*"

Sound clew

Number	Function	User Input
****	Come in sound clew	
100#	Web setup	
110#	DHCP setup	1# start 2# stop # return list
120#	IP address setup	use "*" replace "." . "#" exit sample：172*16*21*10# press "#" return
130#	Gateway setup	use "*" replace "." . "#" exit press "#" return
140#	Network mask setup	use "*" replace "." . "#" exit press "#" return

5.2 Config IP Phone through Web browser

5.2.1 Access to the IP Phone configuration menu

The IP Phone Web Configuration Menu can be accessed by the following URI:
http://Phone-IP-Address. The IP address can be set to either WAN IP address or
LAN IP address, default factory setting of WAN configuration is DHCP Client model, default LAN IP address is "192.168.10.1".
If connect PC with IP Phone LAN port and config to obtain IP address automatically, you could check the default gateway IP namely LAN IP address of IP Phone. The procedure as below

Home LAN SIP CODECS System Download Configuration Reset Logout
Home
Welcome to the VoIP Phone download and configuration utility.
Select from the configuration options in the menu on the top.
System Information
System Uptime: 0 days, 0h 51m 33s
NTP time: 06:19PM 09/16/2006 (GMT+8)
LAN IP Address: 192.168.1.108 (Dynamic)
MAC Address: 00:90:ff:56:77:65
Serial Number:
Security: Password installed
Application Code Version: VR 4.2 (MSCS P20003) Build-Date: Sep 11 2006
Downloader Code Version: VR 4.2 (MSCS P20003)
Copyright 2006.

5.2.2 User verification

Users are requested to make verification when config or browse the IP phone thru web pages, users can direct login the config menu by inputting username and password as below ,

 Guest account: both default username and password are “guest”, enable users to browse IP Phone parameter with low-level user interface

 Administrator account: both default username and password are “admin”, enable users to configurate IP Phone with high-level user interface

5.2.4 Network configuration

Network configuration includes LAN Config.

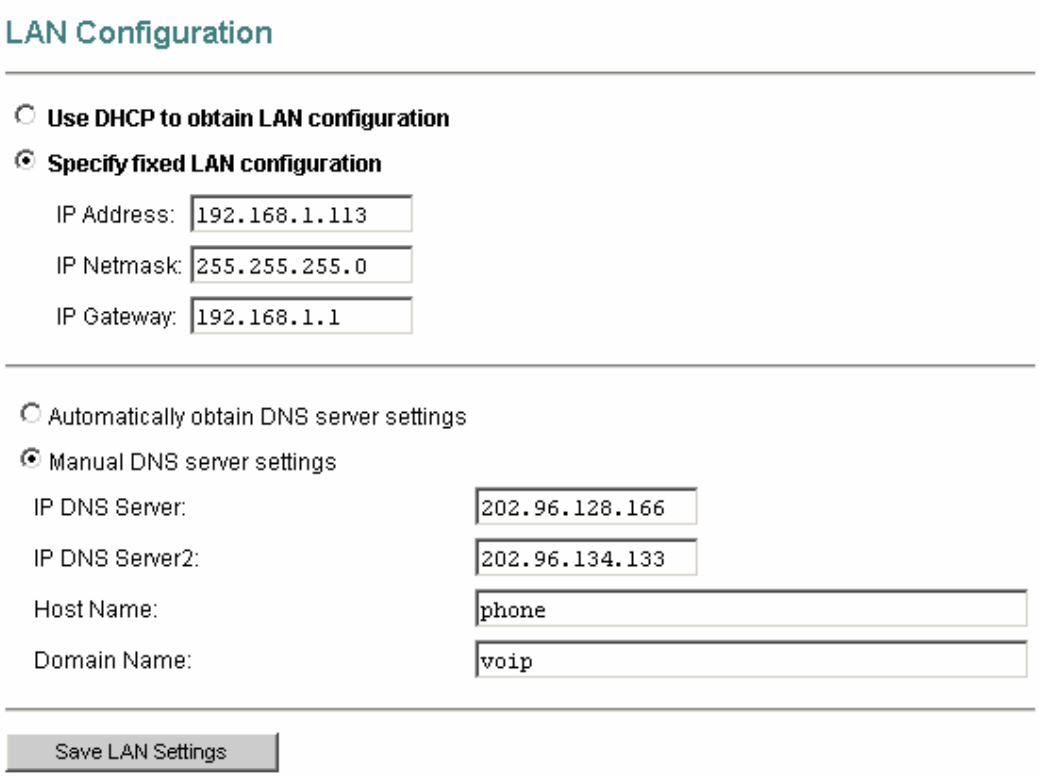
Two models (Static /DHCP) are paratactic. Users can set the right model base on actual requirements.

 Connect network to internet thru Static mode

Default network config is static model; Users need to set below parameters LAN Configuration This web page displays the LAN parameter configuration. Please note once the bridging mode is selected, the LAN configuration will be no longer effective.

Configuration Example

- Config LAN: generally config one private IP address
- Start LAN DHCP Service and NAT or not: default setting is start

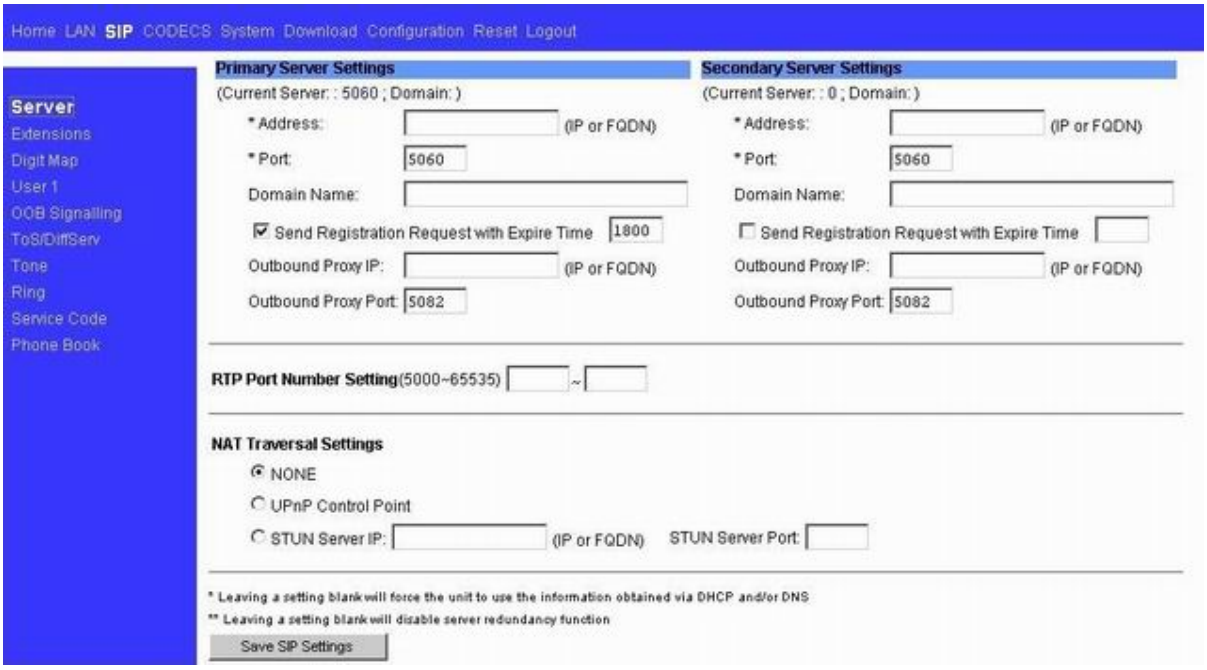


Start Bridge Mode or not（transparent mode）: Once start Bridge Mode, some parts of LAN config will be disabled, and the phone will no longer set IP address for LAN physical port，LAN and WAN will join in the same network；

5.2.3 VoIP Configuration

This section is to config signaling protocol for the SIP Server and Client.

User can configure specific parameter of H323 signaling protocol on this page；



Definition of each parameter described as below

SIP[Unregistered] Configuration	SIP register state ; if register successfully, show“Registered” in the square bracket , otherwise show Unregistered
Register Server address	Set SIP register server IP address
Proxy Server address	Set proxy server IP address (usually SIP will provide the same configuration of proxy server and register server, if different(such as different IP addresses), then each server’s configuration should be modified separately)
Register Server Port	Set SIP register server signal port
Proxy Server Port	Set SIP proxy server signal port
Register Username	Set SIP register server account username (Usually it is the same with the config port number)
Proxy Username	Set the SIP proxy server account username
Register Password	Set password of SIP register server account
Proxy Password	Set password of SIP register account
Domain Realm	Set the domain realm (Usually it is same with registered server and proxy server IP address)
Local SIP Port	Set local signal port, the default is 5060
Phone Number	Set assigned phone number
Register Expire Time	Set expire time of SIP server register, default is 120 seconds
Detect Interval Time	Set detection interval time of server, default is 120 seconds
RFC Protocol Edition	Enable the phone to use protocol edition. When the phone need to communicate with phones using SIP1.0 such as CISCO5300 and so on, need to modify into RFC2543. the default is

	to RFC3261;
DTMF Mode	Set DTMF sending mode
User Agent	Set the user agent if have, default is common
Enable Register	Configure enable/disable register
Auto Detct Server	Set the phone enables to detect server automatically
Enable Pub Outbound Proxy	Configure to enable to use public outbound proxy
Server Auto Swap	Configure main and backup auto-swap server; if the phone enables main and backup server function, the automatic detection and auto-swap functions should both be chosen
SIP (Default Protocol)	Set SIP as the default signaling protocol

After finished the aforesaid network and VoIP configurations on the phone and network communication has been implemented，the user can make VoIP calls by the calling register and proxy server.

Note:

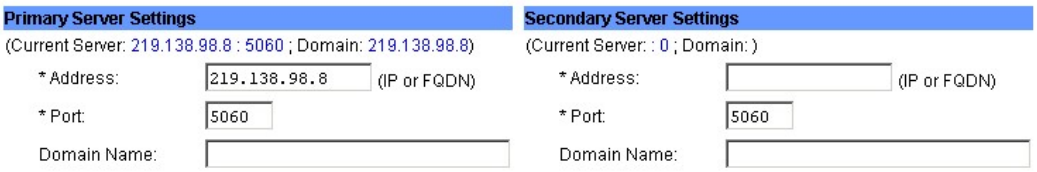
Some ISP internet may inhibit the phone to register and cancel the register in process, so user had better cancel apply or register soon and then submit registration repeatedly. Server may stop response of dialogue machine, then the phone receives no register/cancel login request and registration state will show incorrectness!

5.2.4 Config Manage (Save and Clear Configuration)

Configuration Example

Firstly users should get the account info from VOIP Operator (Including Server IP address, port, username, password etc.) and follow below procedure.

 Config registered server and proxy server IP address and signaling port. (Support DNS for registered server and proxy server)

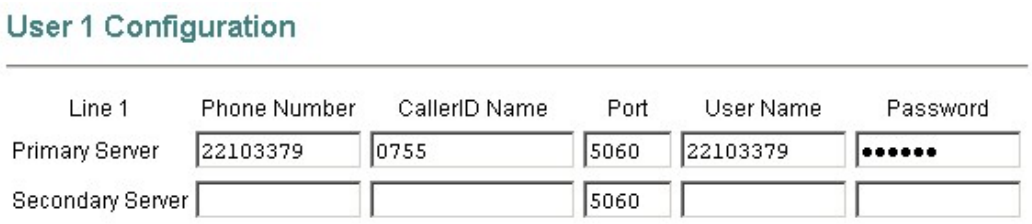

Primary Server Settings
(Current Server: 219.138.98.8 : 5060 ; Domain: 219.138.98.8)
* Address: 219.138.98.8 (IP or FQDN)
* Port: 5060
Domain Name:
Secondary Server Settings
(Current Server: : 0 ; Domain:)
* Address: (IP or FQDN)
* Port: 5060
Domain Name:

* Config the username and password for registered server and proxy server.

* Config the phone number (Usually phone number is same with SIP account)

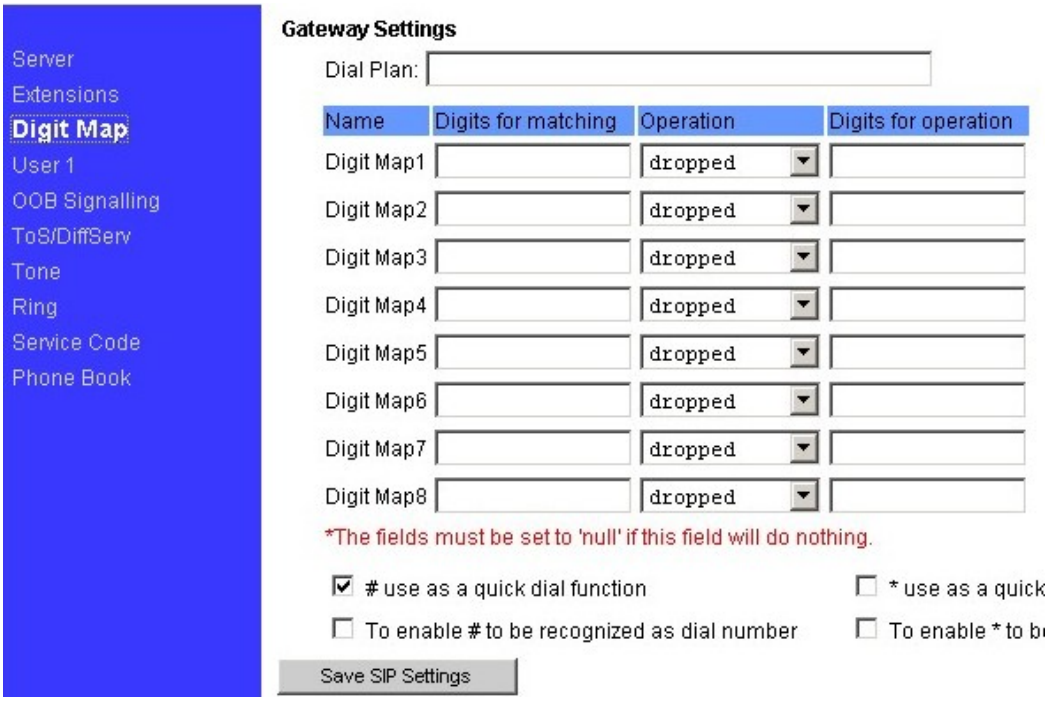
Remark: due to the above register username is “client”, so the phone number is different from SIP account)

Remark: due to the above register username is “client”, so the phone number is different from SIP account)

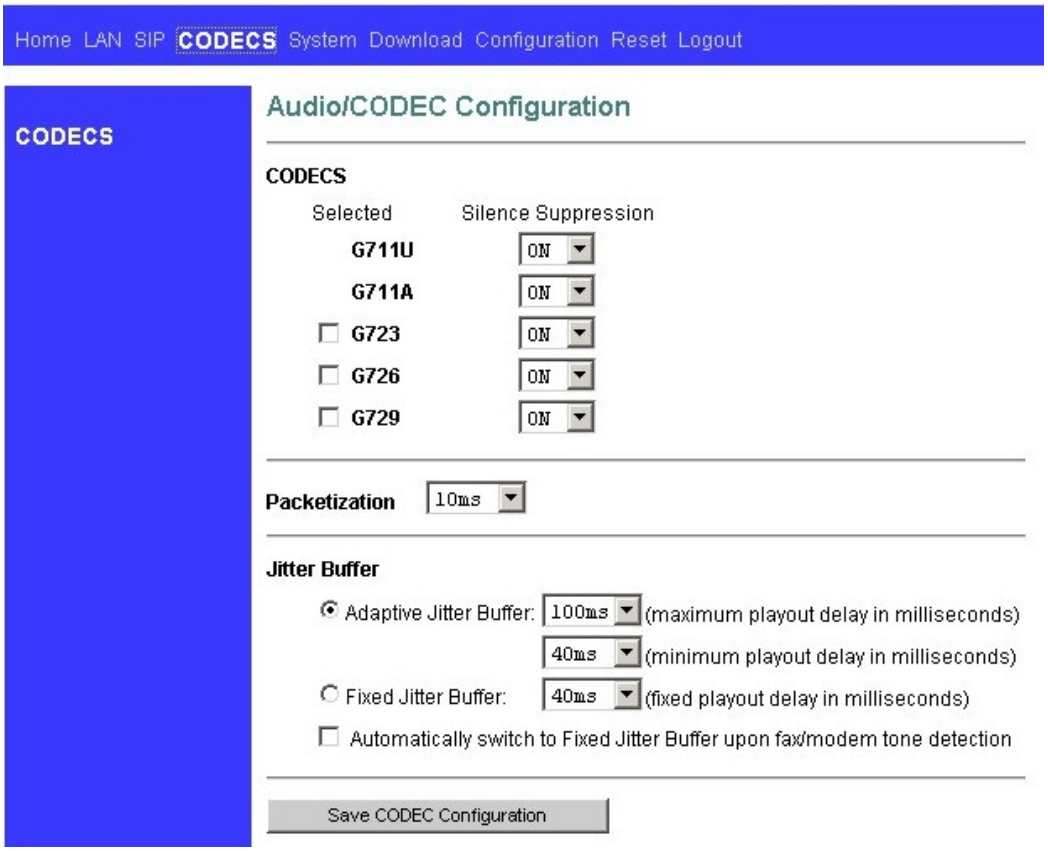


5.2.5. System Reboot

Once any change of phone configuration is made, users need to reset IP phone to go into effect. Users should save the modified configuration before system reboot, otherwise the phone system configuration will go back to last saved setting. The system reboot interface as below



5.2.6 CODECS Setup



5.2.7 System Manage

1. Account Manage

Users can edit users (add or delete) account and modify existing users' authority on this web page.

2. Time Set

Setting time zone and SNTP (Simple Network Time Protocol) server according to users location, users could manual adjust date and time on this web page.

Definition of each parameter described as below

Operation Example

□ Add one new account

Click “Add” button →input User name (No-Modify) →Choosing User level from dropdown menu →set new user password →confirm password account info, return to account configuration interface by clicking “Change Password” button

Owing to the phone's default account： accounts of the administrator level-admin account and the ordinary level－guest account are all weak account and weak password, the username and password will be easily to guess on public network, so the user had better modify the administrator and ordinary user.

Enter with manager level when making modification，create a administrator account and a browse account (you'd better not set the name as admin, administrator, guest, etc.)， set password and then save configuration， entering with new manager account, delete default manager and browse account and save configuration， security will be enhanced!

3. Ringer Tone Set

6.1 Web Update

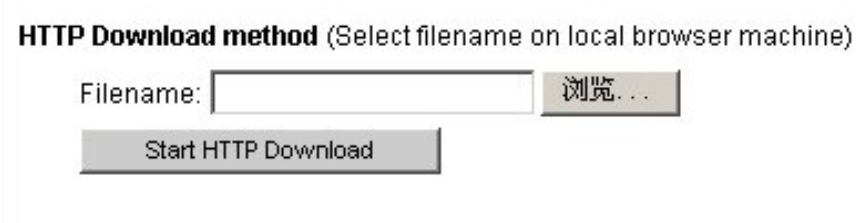
On this page, user can select the upgrade document (firmware or config file) from hard disk of the computer directly to run the system upgrade. After upgrade completed ， reset the phone and it will be usable immediately.

STEP:

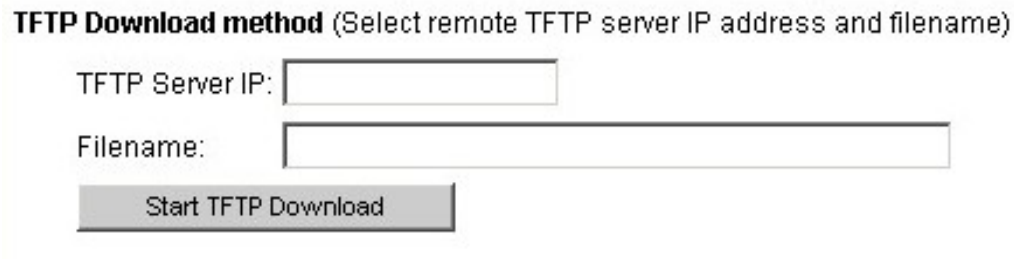
Enter Update menu →WEB Update submenu→ click “browse” button→ download upgrade document from hard disk (firmware or config file provided by manufacturer) → click “Update” button →reboot IP phone to go into effect

Note:

Under system upgrade progress, IP Phone may not be restarted normally due to some system reason (e.g. electricity shut off), users can re-download under protected mode.



6.1FTP or TFTP server before using FTP update way.

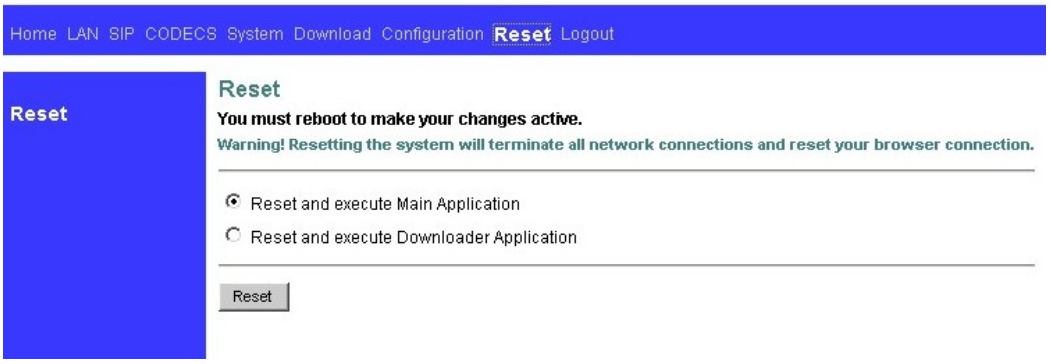


6.2 URL Update way.

Definition of each parameter described as below

6.3 IP Phone System Reboot

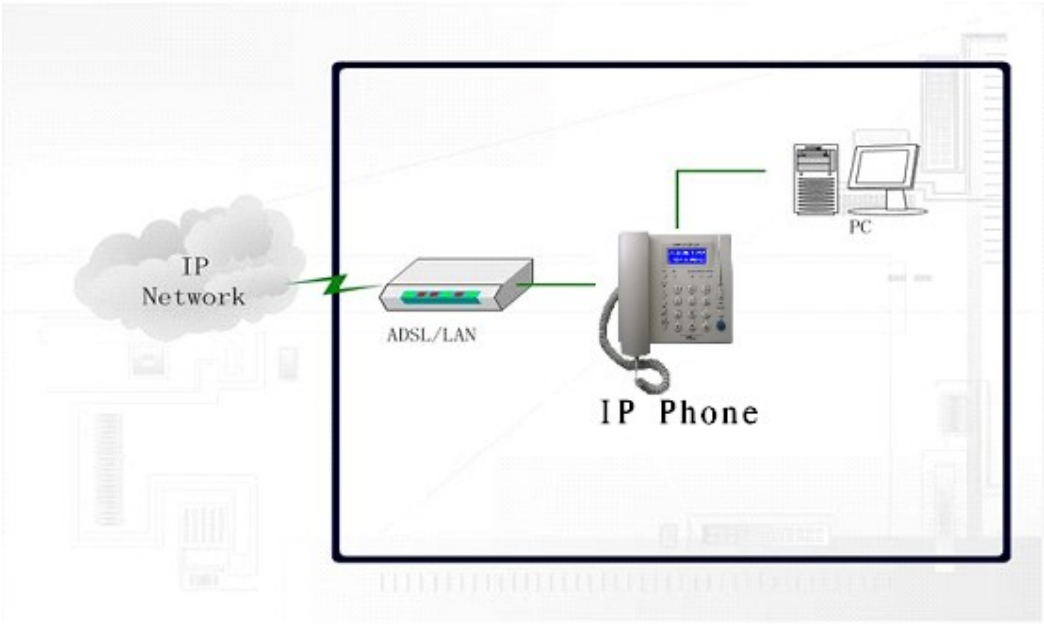
Once any change of phone configuration is made, users need to reset IP phone to go into effect. Users should save the modified configuration before system reboot, otherwise the phone system configuration will go back to last saved setting. The system reboot interface as below

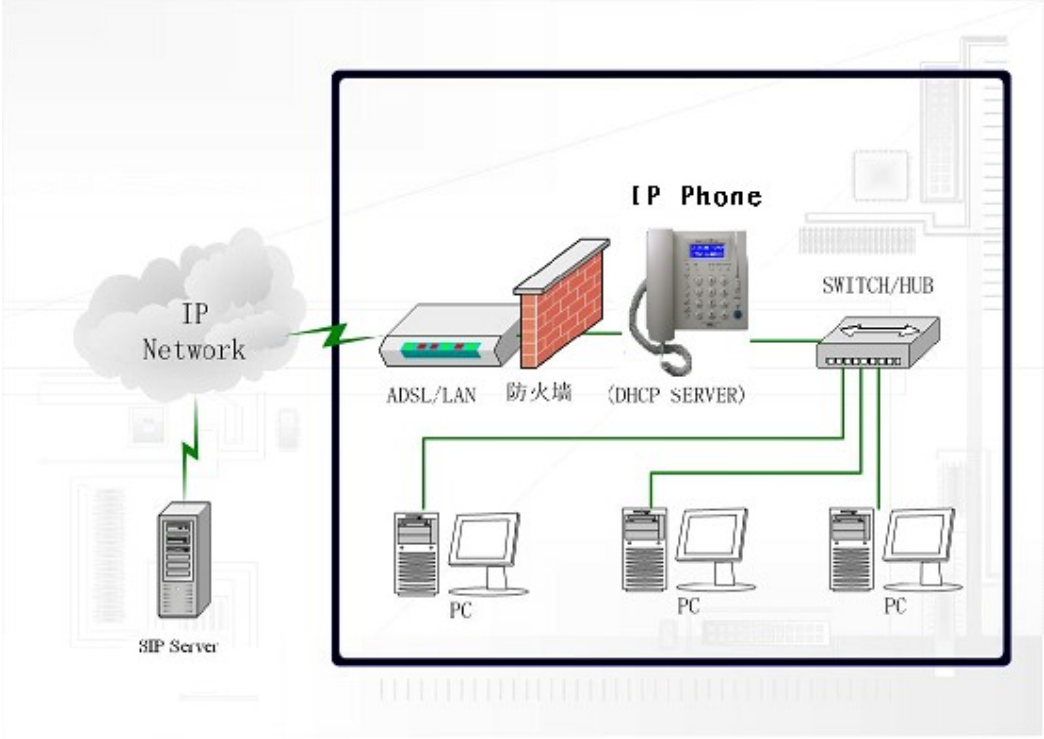

Home LAN SIP CODECS System Download Configuration Reset Logout
Reset
Reset
You must reboot to make your changes active.
Warning! Resetting the system will terminate all network connections and reset your browser connection.
Reset and execute Main Application
Reset and execute Downloader Application
Reset

7. Default Factory Setting

 Gain IP address thru DHCP mode, the static IP address of WAN Port is 192.168.1.176, the IP address of LAN port is 192.168.10.1.Default is to start the DHCP service and NAT function.
 Default communication protocol is to use SIP, SIP port is 5060
 Default HTTP port is 80, Telnet port is 23
 Default number end is "#" button
 Default user account is admin and guest
 Default phone time is to use SNTP protocol to get GMT

8. General System Connect of IP Phone


IP
Network
ADSL/LAN
PC
IP Phone


IP Phone
IP
Network
SWITCH/HUB
ADSL/LAN
防火墙
(DHCP SERVER)
PC
PC
PC
SIP Server